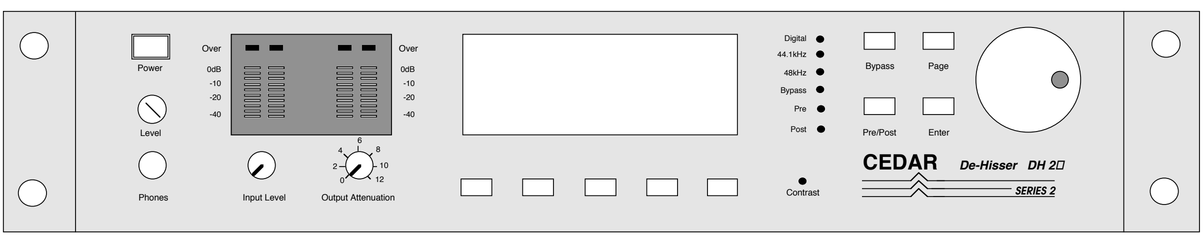
# CEDAR

Professional Hardware Systems

# DH-2 De-Hisser

Digital Audio Restoration System

*SERIES 2*




OWNER'S MANUAL

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# INTRODUCTION

Thank you for purchasing the CEDAR DH-2 De-Hisser Module. This is the world's most advanced dedicated single-ended noise removal unit, and offers processing power and performance that could only previously be obtained using digital signal processors (DSPs) installed in desk-top (or larger) computer systems such as CEDAR for Windows. The De-Hisser is designed for professional use, although it will work perfectly well in a domestic environment, and its features include the following:

- Revolutionary noise removal algorithms
- No need for a “Spectral Fingerprint”
- The latest 'SERIES-2' CEDAR hardware
- Digital Audio interfaces conforming to the AES/EBU and SPDIF standards
- 24-bit input and output resolution when using AES/EBU interfaces
- Three sample rates supported on digital inputs: 32kHz, 44.1kHz and 48kHz
- Two sample rates supported on analogue inputs: 44.1kHz and 48kHz
- Balanced analogue inputs and outputs for connection to professional analogue equipment
- ADC and DAC converters using the latest 64x over-sampling $\Delta$-$\sum$ (Delta-Sigma) technology
- >103dB dynamic range A/D and >93dB dynamic range D/A
- Mountable in a 19" EIA rack
- Remote control via MIDI and RS232 interfaces
- SMPTE/EBU timecode capabilities via optional upgrade
- Input and output LED bar-graph VU meters
- Twin 40-bit floating point DSP processors delivering 50MFlops to handle the most complex audio processing requirements
- High levels of artificial intelligence designed into the DH-2 program algorithms making it extremely simple to use

# THE BACKGROUND TO CEDAR NOISE REMOVAL

Cheap digital audio (i.e. CD) has made discerning listeners quite intolerant of the noises and distortions present in analogue audio signals. After all, in a perfect digital world there are no clicks, crackle, pops, buzzes or hums, and no hiss - so it's a shame that we live in a far from perfect world. Even today, the vast majority of mixing desks still have all-analogue signal paths, so most DDD-classified CDs are still mastered through numerous analogue stages. And 'vintage' (i.e. pre-1980s') recordings are by definition re-mastered from analogue disks and master tapes which inevitably suffer from at least one of the degradations listed above. So recording engineers are turning more and more to the technologies that reduce the noise added in the signal path, or are capable of removing it from the final recording.

Before proceeding any further, perhaps it would be best to describe what we mean by the term 'broadband' noise, defining it to be a random effect which adds (or subtracts) a random amplitude at all times to (or from) all frequencies within the audio spectrum. Thus, we do not include artefacts of very limited duration such as intermittent electrical clicks or microphone 'grounding' which may be corrected by quite different methods to those described below.

Next, let's dispel any illusions regarding Dolby, dbx, and similar noise reduction systems. These are dual-ended processes designed to minimise the accumulation of any extra noise added by the limitations of analogue recording tape. (Dual-ended processes are commonly called encode/decode systems because the recording process 'encodes', and the playback process 'decodes', the signal.) None of these processes enable you to remove noise from within a signal that already contains it - they simply stop you adding too much more when you commit that signal to tape and then play it back again. Perversely, they therefore help your tape deck to accurately record, and then faithfully reproduce, any noise contained in the original signal. So, what you need is a 'single-ended' process that removes noise from your signals prior to committing them to tape, or at the very least, improves the signal to noise (S/N) ratio without affecting the signal adversely. Which brings us neatly to the volume control... stunningly effective at removing noise, it does nothing to improve the S/N ratio, and has an all-too-noticeable side-effect. No noise, No signal.

Since broadband noise is most intrusive at high frequencies, where the masking effect of loud sounds is least present, the first stage in our evolutionary tale of noise removal is the simple low-pass filter. Less damaging than the volume control which removes the signal altogether, this only removes a proportion of any signal present above its cut-off frequency, $f_C$. Unfortunately, if, at the given frequency, you reduce the amplitude of the noise content of your recording by, say, 6dB, you also reduce the genuine signal at this frequency by the same amount. This may be acceptable if the recording has little or no high frequency content, but natural sounds and modern electronic instruments have significant components up to and beyond the limits of human hearing. Consequently, the low-pass filter will only be successful in processing your antique collection of '78's, and even then only at a cost.

But this gives us a hint as to how a more effective single-ended noise reduction system could be designed: perhaps a device could be built which removes the high frequencies when there is no signal present, but leaves them untouched when the noise is being masked by genuine high frequencies? It's a Dynamic Filter (so called because $f_C$ moves dynamically according to the signal content). But such devices are limited: for one thing, they can only remove the noise which exists above $f_C$, which is itself an inaccurate representation of the highest frequencies contained in the genuine signal at any given time. Secondly, and in common with the simple low-pass filter, they have roll-offs typically of the order -12dB/octave or -6dB/octave, so they always allow some high frequencies through. And thirdly, even though the filters are designed to track the signal very quickly, they cannot respond instantaneously, so they tend to round off fast transients. And, because their *raison d'être* is to reduce the signal bandwidth they also tend to dull the genuine signal quite perceptibly. So to summarise dynamic filters: if you're not compromising the signal you may not be removing as much noise as you wish, and if you're removing all the noise you're probably damaging the genuine signal.

Perhaps an alternative approach could give better results? Instead of altering the frequency response of the signal to reduce the noise content, how about changing the overall signal level in some way? This isn't such a strange idea. Consider: if noise of a relatively constant amplitude is always present in a signal then, if the total signal amplitude drops down to the noise level, we can assume that no genuine signal is present. While there are many flaws in this argument (largely to do with the statistical nature of broadband noise) it suggests a device which will eliminate some of the noise: a Noise Gate. This detects when the signal drops below a certain 'threshold' set by the user and then cuts off the signal entirely. There are many enhancements to the Gate idea, such as variable attack and release times, and hysteresis (all added to limit the occurrence of damaging side-effects) but the principle always remains the same: if the total signal drops below the amplitude threshold, the gate shuts off the signal.

Just as the filter can be improved by making it dynamic, so can the gate. Such a device is called an Expander. The Expander still has a threshold control, but unlike the gate, the Expander applies a progressive gain reduction, the amount of which is determined by the settings selected by the user. For example, if a signal drops 3dB below the threshold, the Expander may reduce the signal volume by 6dB, 12dB, or any other figure, depending upon the expansion ratio. Unfortunately, the subjective difference between the noise gate and the expander is small and, though they sound great in theory, they don't sound so great in reality. Being time domain processes, they have advantages and disadvantages when compared to frequency domain filtering, but only sound different, not better. In particular, they have the distinct advantage that, when active, they remove all the noise, but once inactive (or ‘open’) no noise is removed. And if you adjust the threshold so that noise can only come through when the signal is loud enough to mask it, you lose the ability to include quiet passages in your recordings.

Commercial noise reduction units now feature a combination of dynamic filtering, expansion, and even compression and excitation - effects which have been included in an attempt to obscure some of the undesirable side-effects of the noise reduction processes. But these are only partially successful, and you still can't master full bandwidth CDs or film soundtracks with them. The results simply aren't good enough.

Whenever a single-band expander encounters a signal with amplitude below its threshold, it further reduces the volume. But what if there is still a significant signal at (say) 3kHz, but very little elsewhere in the frequency spectrum? The single-band expander has no way of discerning the genuine signal, and shuts this out at the same time as all other frequencies. A multi-band device separates the audio spectrum into a number of bands, and treats each of these as an individual signal. This form of noise reduction has evolved from simple analogue units onto Digital Audio Workstations which can incorporate high power DSPs capable of splitting the audio spectrum into multiple bands and applying expansion to each of these in the digital domain. But multi-band units are still unable to distinguish accurately between genuine signal and noise. They still act upon the inaccurate assumption that, if the total signal level approaches its noise floor, all that is present is broadband noise. Consequently, even the most sophisticated downwards expanders and dynamic filters remove some of the genuine signal. Furthermore, the poor band separation filters (typically of the order -6dB/oct or -12dB/oct) severely limit performance. The consequences of these problems are well understood and largely unavoidable: loss of high frequencies, loss of ambience, and degradation of hard transients.

All the methods so far described use filters, gain controls, or a combination of both to achieve their results. Whether implemented in either the analogue or digital domain, all such filters and gain controls are 'ratio' devices - that is, if (at any given frequency) you remove half the power of the noise, you remove half the power of the signal; if you remove 3/4 of the noise, you remove 3/4 of the signal... and so on. Consider now a signal that has, at a given frequency, 100 units of 'volume' on some arbitrary scale. By measuring the noise content of that signal during an otherwise silent passage, you can determine that there are, say, 20 units of noise present at that frequency. It should be possible to remove this noise by removing 20% of the signal. But what if, a moment later, the total ‘volume' of the signal drops to 40 units? An analogue filter, removing 20% of the signal in the same manner, will only remove 8 units of the 20. On the other hand, a subtractive filter (which is practical only in the digital domain) will still remove the full 20 units, equivalent to a filter reduction of 50%. This is, of course, what we want, because the noise at this moment represents 50% of the total signal amplitude. So now we arrive at the most sophisticated noise removal technology yet implemented in a commercial sense: computerised spectral subtraction.

The concept of Spectral Subtraction becomes useful when a DSP is used to split an audio signal into hundreds of bands. You can then be very precise about how much noise you remove, subtracting a lot at (say) 8kHz, while leaving 8.1kHz virtually untouched. But if this sounds to good to be true, it is. The noise spectrum (the sonic 'fingerprint') can only be accurately measured if there is an otherwise silent passage within the music. If the fingerprint is not accurate the amounts subtracted will be wrong, leading to unpleasant side-effects. And, worse still, many tracks are 'close edited', making it impossible to obtain any such fingerprint. But let's assume that you have obtained a perfect fingerprint. You might expect to produce a good restoration, with large amounts of noise removed and few or no side-effects. Yet experience shows that all attempts to implement spectral subtraction in this unmodified form produce both (i) unacceptable artefacts, and (ii) unusably dry and dull results. This is, in part, because the fingerprint is merely a snapshot of the random noise, accurate only at the instant at which it is taken. Because the noise content is constantly changing, an unmodified subtractive algorithm will be deriving its result from inappropriate data. Garbage in, garbage out.

These pitfalls have prompted much commercial and academic research, and many companies and independent researchers have investigated enhancements designed to overcome them. CEDAR Audio's developments are embodied in its product, 'HISS2', which incorporates an algorithm that updates the noise fingerprint every 1,000 samples. This allows the system to track variations in the noise content. It also features algorithms that prevent the compression of incoming transients, and that distinguish between true noise and, for example, reverberation tails of the genuine signal. These and other features avoid many of the pitfalls of simple spectral subtraction, and allow users to remove noise without undue damage to the genuine signal contained within the source.

But this system still requires a noise fingerprint, whether captured from the signal or created by arbitrary means. Consequently, the user-interface is complex, and the system cannot be implemented in a stand-alone box. The greatest simplification of this interface requires an algorithm capable of autonomous determination of the noise content. Such methods are the subject of much on-going research although, as yet, only two have been proved sufficiently accurate to be incorporated within commercial products. Developed by CEDAR, these algorithms make possible a stand-alone module that dispenses with the user-derived noise fingerprint. The algorithm will itself analyse the noise content of the signal and apply an automated noise reduction process. Further development of this process has led to an enhanced computer-based software module called Dehiss2, which may be found in the CEDAR for Windows™ suite of audio restoration products, as well as the latest DH-2 De-Hisser.

*Dolby and dbx are trademarks of their respective manufacturers.*

# SAFETY INSTRUCTIONS

## CAUTION:

1. **Read all of these instructions**
   All safety and operating instructions should be read before the DH-2 is operated.

2. **Save these instructions for future reference.**

3. **Follow all warnings and instructions.**

4. **Water and Moisture**
   The DH-2 should not be used near water, and must not be exposed to rain or moisture. If the DH-2 is brought directly from a cold environment into a warm one, moisture may condense inside the unit. This, in itself, will not damage the DH-2, but may cause hazardous electrical shorting to occur. This could severely damage the DH-2, and even cause danger to life. ALWAYS allow time for the DH-2 to naturally reach ambient temperatures before connecting the mains power.

5. **Mounting**
   The DH-2 should be carefully mounted in a 19" EIA rack, or placed on a flat, stable surface. If it is used on a cart or free stand, care should be taken when it is moved: uneven surfaces or excessive force may cause cart and DH-2 to overturn. Do not position the DH-2 in a place subject to strong sunlight, excessive dust, mechanical vibration or periodic shocks.

6. **Wall or Ceiling Mounting**
   The DH-2 has not been designed for mounting directly to walls or ceilings.

7. **Ventilation**
   Good air circulation is essential to prevent internal heat build-up within the DH-2. The DH-2 should be situated so that its position does not interfere with proper ventilation. The DH-2 should not be placed in any situation which impedes the flow of air through the vents at the front and rear. Do not place the DH-2 on a soft surface.

8. **External Heat Sources**
   The DH-2 should be installed away from significant heat sources such as radiators, and (if possible) away from other audio devices such as amplifiers that produce large amounts of heat. Installation in racks with devices such as signal processors or tape machines should not be a problem.

9. **Power Sources**
   The DH-2 features an auto-switching power supply which will work safely on any mains supply in the ranges 95v/130v and 190v/260v, 50Hz or 60Hz AC only.

   **You should never attempt to modify or adjust the internal power supply in any way. It contains no user serviceable parts.**

10. **Grounding or Polarisation**
The DH-2 should always be grounded (or 'earthed').

11. **Power Cord Protection**
Power connectors should be routed so that they will not be walked on or pinched.

12. **Extended Periods of Non-Use**
The DH-2 is not disconnected from the mains power as long as it is connected to the wall outlet, even if the unit itself has been switched off. Therefore, if the DH-2 is not to be used for an extended period of time, unplug the unit from the wall. Pull the connector out by the plug, never by the cord itself.

13. **Cleaning**
Clean only with a dry cloth. NEVER use liquid cleaners such as alcohol or benzene on the DH-2. NEVER use abrasive pads on the DH-2.

14. **Damage Requiring Service**
The DH-2 should be returned to qualified service personnel when:
- objects have fallen into the unit
- liquid has been spilled into the unit
- the unit has been exposed to rain
- the unit fails to function or appears to operate abnormally
- the unit has been dropped, or the case damaged.

15. **Servicing**
The user should not attempt to service the DH-2 beyond the instructions contained in the User's Manual. All other servicing should be referred to qualified service personnel.

# SET UP

## 1. UNPACKING AND INSPECTION

Be careful not to damage the DH-2 during unpacking. Save the carton and all packing materials since you may need them to transport the DH-2 in the future.

In addition to the DH-2 and the packaging, the carton should contain the following:

- mains connection lead
- this manual
- blanking plates which may be used to replace the rack-mount ears
- DH-2 Tutorial DAT

## 2. INSTALLATION SITE

The DH-2 may be used in most areas, but to maintain reliability and prolong operating life observe the following environmental considerations:

- Nominal temperature should be maintained between 5˚ and 35˚ Centigrade (41˚ and 95˚ Fahrenheit).

- Relative humidity should be in the range 30% to 60% non-condensing.

- Strong magnetic fields should not exist nearby.

## 3. RACK MOUNTING

The DH-2 can be mounted in a standard 19" EIA rack.

## 4. FREE STANDING USE

The DH-2 can be used as a free-standing unit. The rack-mount ears may then be replaced by the blanking plates if desired.

To replace the ears with the blanking plates:

- Unscrew the three bolts which attach each ear to the chassis of the DH-2.

- Attach the blanking plates using the same retaining bolts. Do not over-tighten these bolts as doing so may cause damage to the DH-2.

# CONNECTIONS

The DH-2 may be connected to most of the professional audio equipment currently available. Three types of audio input and output are provided (one analogue and two digital) and these will satisfy most users' interconnection requirements. Full descriptions of these connectors will be found later in the manual.

## 1. BEFORE CONNECTION

- To prevent problems and possible equipment damage, turn off the power to all equipment before making connections.
- Be sure to insert plugs firmly into sockets. Loose connections may cause hum and noise.
- When unplugging any lead, do so by grasping the plug, not the lead.

## 2. POWER CONNECTIONS

Ensure that the DH-2 is switched OFF before inserting the mains lead.

***NOTE: Users with 2-pin mains supplies:***
*When the DH-2 is connected to other audio components, the AC hum of the unit may be increased or decreased by reversing the direction of the power connector in the socket. Check that the cord is in the favourable position ('in-phase') with respect to other audio devices in the chain. This will ensure that the best sound quality is obtained from your DH-2.*

*For further information on grounding and polarity consult a person familiar with studio grounding techniques.*

## 3. SIGNAL LEAD CONNECTIONS

Refer to the Rear Panel diagram:

The DH-2 offers three audio connection standards: one analogue and two digital. These are:

- balanced analogue audio I/O
- digital SPDIF format audio data
- digital AES/EBU format audio data

*Note that the DH-2 always passes its output to all three signal outputs irrespective of the input used, but that the digital data will only be formatted for* ***either*** *AES/EBU* ***or*** *SPDIF, as defined by the user parameters.*

**(i) Balanced analogue audio I/O (Pin 2 - 'hot')**

This standard is used in professional audio equipment. Connect the output from your source to the balanced analogue inputs of the DH-2 using standard XLR plugs. You will require two such connections: one for each channel.

The balanced audio output may be used to connect the DH-2 directly to audio equipment such as mixing desks and professional recorders featuring balanced XLR inputs and outputs.

**(ii) Digital SPDIF format audio data**

The SPDIF format is used by domestic and semi-professional digital audio devices such as DAT machines, some ADCs, and some CD players. Both audio channels are carried along a single cable, so you may connect the SPDIF output from your source to the SPDIF input of the DH-2 using a single cable terminated with RCA (or 'phono') plugs.

The SPDIF output of the DH-2 may be connected to the SPDIF input of your recording device or external DAC.

**(iii) Digital AES/EBU format audio data**

The digital AES/EBU format is used by professional digital audio devices including mastering systems, DASH recorders, and high quality ADCs & DACs. Both channels of audio are carried along a single cable, so you may connect the AES/EBU output from your source to the AES/EBU input of the DH-2 using a single cable terminated with XLR plugs.

The AES/EBU output of the DH-2 may be connected to the AES/EBU input of your digital mixer, recording device or external DAC.

**24-bit Digital data resolution:**

The DH-2 features 24-bit input and output resolution whenever the AES/EBU digital input and output are utilised.

**Dithering:**

The DH-2 also features TPDF (Triangular Probability Density Function) dithering. This is applied to the digital data when the SPDIF output format is selected. Dithering is always applied to the data presented to the DACs.

*Note: In order to fully comply with EMC regulations, this unit should be connected via its AES/EBU and/or analogue connectors. Metal-shelled XLR connectors should be used. We recommend using a good quality 'starquad' cable, with three cores connected to pins 1, 2 & 3. The shield of the cable should be connected, at both ends, to the outer shell of the connector.*

# 4. OTHER CONNECTIONS

**(i) SMPTE/EBU**

The DH-2 has been designed to host an optional SMPTE/EBU interface offering LTC and VITC protocols. The standard DH-2 does not support timecode and these connectors are not present.

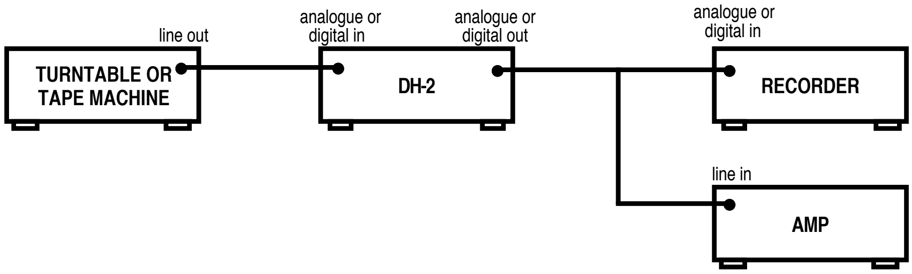
**(ii) MIDI IN/OUT/THRU**

The operation of the DH-2 may be controlled using the Musical Instrument Digital Interface (MIDI). Refer to the chapter on Remote Control Protocols for further instructions.

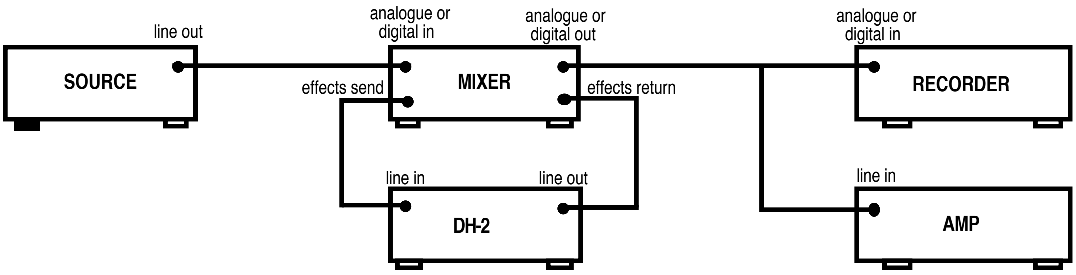
**(iii) RS232**

The DH-2 may be controlled using the standard RS232 serial communications protocol. Refer to the chapter on Remote Control Protocols for further instructions.

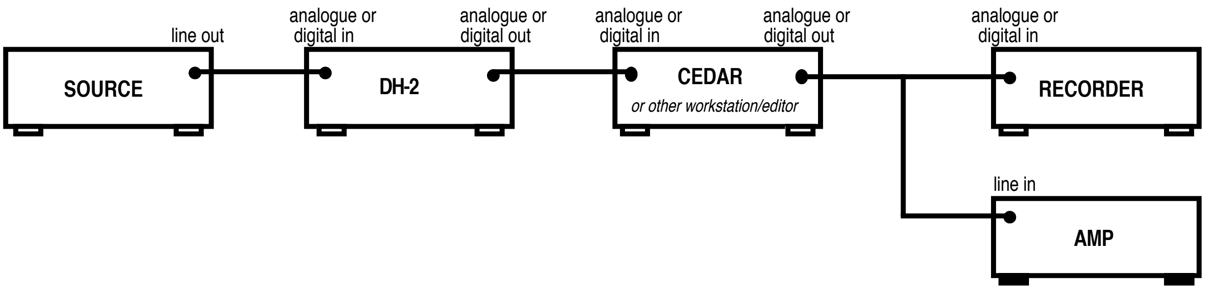
# SAMPLE INSTALLATION IDEAS




1. *DH-2 used in-line for transcription or broadcast purposes.*




2. *DH-2 used on the effects loop within a studio environment.*




3. *DH-2 used in-line prior to an editor or audio workstation.*

# A GUIDE TO RESTORATION PROCESSING

Contrary to ‘common sense’, the order in which restoration processes are carried out makes a great deal of difference to the quality of the final result. Consequently, there is one ‘right way’ and many ‘wrong ways’ to restore your material.

Following these guidelines will help you to achieve the best results on most material:

- De-Clicking (De-Scratching) should ALWAYS be carried out first. This is because:

  i Large clicks make it difficult for the De-Crackling process to identify and remove the tiny clicks and crackles that constitute surface noise, buzz, and other such problems.

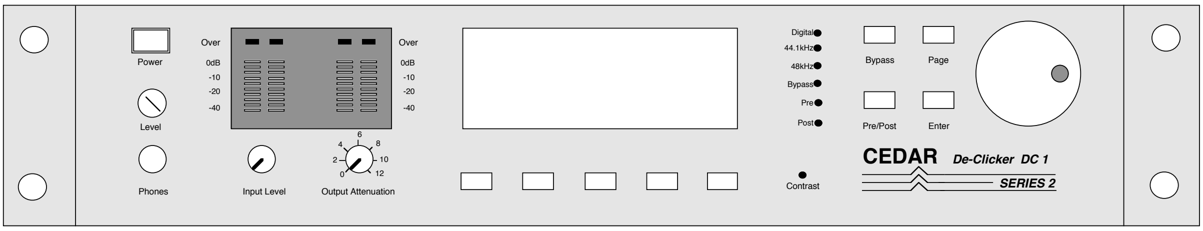
  i i All clicks and scratches are, in effect, tightly defined packets of white noise. If clicks are presented to any of the CEDAR De-Hiss products (HISS-1, HISS-2, Auto De-Hiss, Dehiss, Dehiss2, the DH-1 or the DH-2) they confuse the processes, and create unmusical side-effects. In addition, De-Hissing at this stage will make it almost impossible to identify and remove clicks and scratches at a later time.

- De-Crackling should be the next process because even small crackles can cause the same problems as in (ii) above.

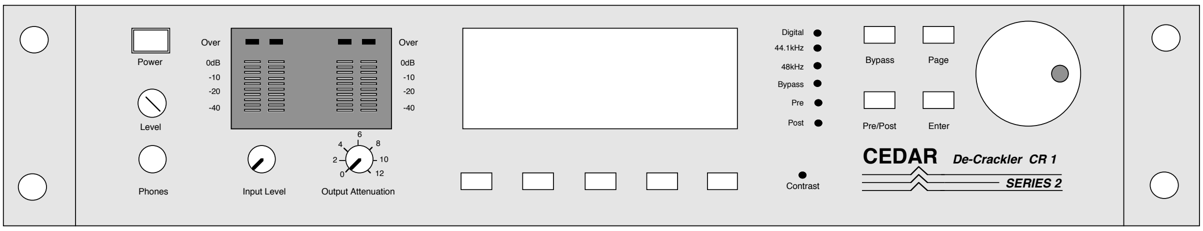
- Azimuth Correction can be carried out either before or after De-Hissing, but experience shows that best results are obtained using the AZ-1 or Phase-EX module before De-Hiss.

- Finally, apply the DH-2.

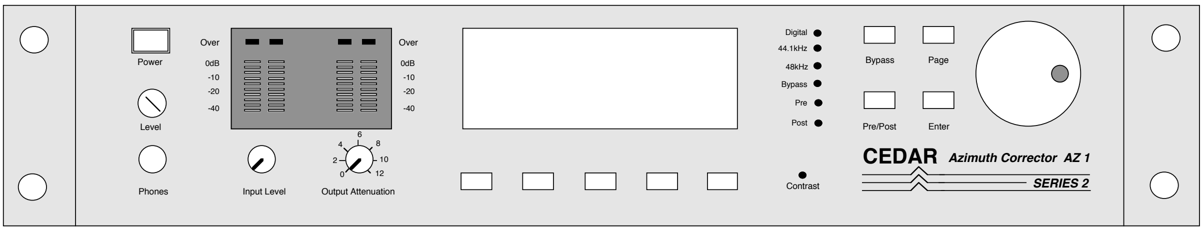
*Note: If you have the full range of CEDAR restoration modules they should be connected as shown in the diagram overleaf. Please note that, to maintain the maximum fidelity and remove and possible sources of degradation between processes, connections between modules should be by AES/EBU (24-bit) format.*



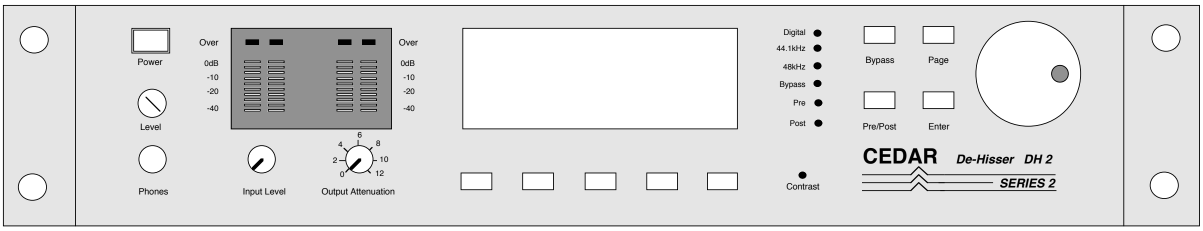

*Firstly, De-Click your material*




*Next, remove crackle and buzz, and reduce distortion if appropriate*




*Then apply Azimuth Correction to material with phase and balance problems*




*Finally, apply noise reduction.*

# LOCATION AND FUNCTION OF FRONT PANEL INDICATORS AND CONTROLS

Refer to the Front Panel diagram:

1. **Power Switch**

2. **Input Signal Meters (Left and Right)**
   Digital signal meters display the peak value of the selected input in dB0s.

   The 'Over' indicators will light if the input signal remains at full scale for four or more consecutive samples.

3. **Output Signal Meters (Left and Right)**
   Calibrated signal meters display the peak value of all output signals.

   The 'Over' indicators will light if the output signal remains at full scale for four or more consecutive samples.

4. **LCD Screen**
   Provides you with a variety of information and messages, keeping you aware of what is currently happening in the DH-2.

   All the control screens of the DH-2 are displayed on the LCD screen. Please refer to the following chapters for full instructions.

5. **Status Indicators**
   Indicate the status of the analogue and digital inputs, and whether the DH-2 is in idle or processing modes.

   Also indicate the possible causes should the unit fail to function.

6. **Dedicated Function Keys.**
   Certain functions are fundamental to operating the DH-2, and these are controlled by the dedicated function keys: Bypass, Page, Pre/Post, and Enter.

7. **$\alpha$-dial (Spinwheel)**
   The $\alpha$-dial enables you to increase and decrease control values. Please refer to the following chapters for full instructions.

8. **Headphone Socket**
   For use with stereo headphones only. Accepts a standard 1/4" stereo jack plug. DO NOT use 2-conductor mono headphones with the DH-2.

9. **Headphone Level Control**
   Use this to adjust for a satisfactory listening level. This level control will not alter the signal level at any of the rear panel outputs.

10. **Input Level Control**
This control acts upon the analogue inputs only. Use it to adjust the volume of incoming analogue signals to the desired level. A level of approximately 0 to -3dB (as shown on the Input Signal Meters) will offer best results.

*The Input Level Control may be physically bypassed internally to obtain the best possible signal to noise ratio (S/N) from the ADCs. This work must be carried out by qualified service personnel, so please refer to your authorised dealer or directly to CEDAR Audio to have this modification performed.*

11. **Output Attenuation Control**
A digital gain control with range 0 to -10dB in 1dB steps.

12. **Function Keys**
Use along with the LCD screen. Please refer to the following chapters for full instructions.

13. **Contrast Control**
The LCD screen may be adjusted for optimum visibility. Use a fine screwdriver to make such adjustments.

# QUICK TOUR

If you are impatient to hear some immediate results using your DH-2 the following instructions should have you up and running within a few minutes:

1. **READ THE SAFETY INSTRUCTIONS.**

2. Connect the DH-2 to the mains supply.

3. Connect your input and output devices to the DH-2 using the appropriate input and output sockets. (If in doubt, please refer to the section CONNECTING THE DH-2 and the manuals of your other equipment).

4. Referring to the front panel diagram, hold down the function key F1 and switch on the DH-2.

5(i) If you are using analogue inputs press PAGE once. Press B (function key F2) to select 'analogue'. Then press PAGE twice more to return to the Control Page.

5(ii) If you are using digital inputs from a consumer format machine such as a domestic DAT recorder press PAGE once, then press B twice to select 'SPDIF'. If you are outputting to a consumer format machine such as a low-cost DAT recorder press A (function key F1) to select SPDIF format.

Press PAGE twice to return to the Control Page

*Note: The DH-2 defaults to AES/EBU PROFESSIONAL format, so skip both instructions 5(i) and 5(ii) if your DH-2 is connected via its AES/EBU input and output.*

6. Play your material through the DH-2.

7. Press function key F2 to select the LEVEL control, and use the $\alpha$-dial to vary LEVEL between 0.00 and 99.00. Provided that the material you are playing contains hiss you will, at some point within the scale, hear it disappear.

8. With ATTEN set to -40.0 and BRT set to 0.00 you will almost certainly hear side-effects while you adjust LEVEL. Experiment with ATTEN and BRT to hear how these affect the output. Please refer to the TUTORIAL section for a full explanation of these controls, how they interact, and how to get the best results from them.

This section should have whetted your appetite, so you should now proceed to the rest of the manual.

# WARMSTART AND COLDSTART

The DH-2 features Warmstart and Coldstart options. Warmstart has been added so that the unit can be configured once, and these parameters are then automatically recalled on every power-up. This is ideal for applications where time-consuming set-ups at the start of each session are not practical.

### Coldstart

If the DH-2 has not been used for some time the system will automatically Coldstart. This process initialises all parameters to their factory default values, and after a few seconds the DH-2 will automatically enter at Page 1.

On start-up the message 'Coldstart' will be displayed at the top right of the start-up screen on the LCD display. The screen will then enter PAGE 1, which will show the default parameters:

| | | | |
|---|---|---|---|
| The default values are: | LEVEL | = | 0.00 |
| | ATTEN | = | -40.0 |
| | BRT | = | 0.00 |

| | | | |
|---|---|---|---|
| Other default values are: | Digital Output | = | AES/EBU |
| | Input Source | = | AES/EBU |
| | Receiver Error Level | = | 1 - Lock |
| | MIDI | = | Channel 1 |
| | Bypass | = | OFF |
| | A to D frequency | = | 44.1kHz |
| | Pre/Post | = | Post |

### Warmstart

The DH-2 remembers the latest parameters used, and the page that was active at the time that the system was last switched off.

On start-up the DH-2 will display the message 'Warmstart' on the screen, and after a few seconds will re-enter at the appropriate page, with all user parameters set to their previous values.

### User Coldstart

If you wish to force the DH-2 to Coldstart, hold down Function Key F1 while switching on the system. Release F1 when the message Coldstart is seen on the LCD display.

*Note: In common with all other digital devices, and irrespective of whether you are Warmstarting or Coldstarting the DH-2, you should always allow a few seconds between switching the unit **off**, and switching it **on** again.*

# OPERATING THE CEDAR DH-2

## 1. DEDICATED CONTROLS:

The DH-2 features a number of dedicated controls to speed operation. These are now explained in turn:

### Bypass

You may wish to bypass completely the operation of the DH-2. Press BYPASS to do this. The current status will be indicated on the Status LED. The Bypass does not 'hard-wire' the input to the output. Analogue signals still pass through the A/D and D/A stages.

*Notes:*

- *There is a delay of approximately 1.3mS in any analogue-to-analogue signal passed through the DH-2 in Bypass mode.*
- *There is a delay of approximately 0.1mS in any digital-to-digital signal passed through the DH-2 in Bypass mode.*
- *All delays are 'group delays' (i.e. are constant at all frequencies) and are measured at a sample rate of 44.1kHz.*

### Page

Use this Function Key to move between Pages.

### Pre/Post

It will often be useful to compare the original signal with the post-processing output of the DH-2. The current status will be indicated on the Status LEDs.

### Enter

The ENTER Key has three functions: as a LOCK-OUT key, preventing accidental changing of parameters; as a CLEAR key, resetting error messages, and as a MIDI DUMP command.

These first two functions are, of course, context sensitive, and the key's action will be appropriate to the page displayed (see below). The MIDI DUMP will be initiated every time that the ENTER key is pressed, regardless of context.

### Input Level

This control acts upon the analogue inputs only. Use it to adjust the volume of incoming signals to the desired level. We recommend a peak level of approximately 0 to -3dB as shown on the Input Signal Meters.

## Output Attenuation

Avoid clipping using the Output Attenuation Control. This is not a compressor or limiter, and acts purely as a digital gain control with variable gain from 0dB to -10dB in 1dB steps.

# OPERATING THE CEDAR DH-2

## 2. PAGES:

The DH-2 has three 'pages' which control all aspects of its operation. Each page is selected by pressing the dedicated **PAGE** function key, displayed on the LCD screen, and may be controlled using the Function Keys and the $\alpha$-dial.

*Note: There is a fourth, normally hidden, page called the Status Page. This is not accessed using the standard 'Page' function, and will be discussed separately in the section describing Error Levels.*

## PAGE 1: CONTROL PAGE

If necessary, access this page by pressing the Dedicated Function Key PAGE until the Control Page appears.

There are five controls in the Control page. These correspond to the five 'soft-keys' and are to be found directly above each of them as follows:

F1 • Stereo Ganging/Left/Right Control
F2 • LEVEL Control
F3 • ATTEN (Attenuation) Control
F4 • BRT (Brightness) Control
F5 • Clear

The controls, and therefore the DH-2 itself, act in the following manner:

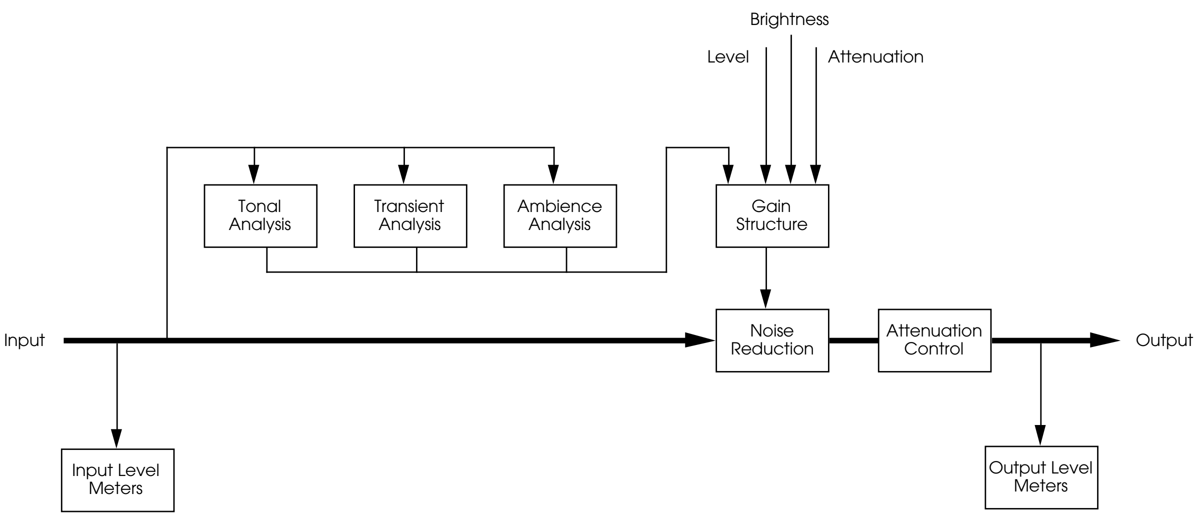



*Figure 1: The DH-2 process overview, and the stages at which the CONTROL PAGE controls modify the signal.*

You can optimise the beneficial effect of the DH-2 by setting each of these controls appropriately. They are now described in turn:

**LEVEL:**

The LEVEL control is used to give the DH-2 algorithm a rough idea of the amount of noise present in any given signal. This is the most sensitive and important control on the DH-2, and incorrect use will result in sub-standard results and/or unwanted side-effects.

- LEVEL = too low
  The DH-2 will not remove all the noise and (especially at high brightness levels) will generate an artefact from residual noise pulses let through by the process. This is often described as 'twittering'.

- LEVEL = too high
  Some low-level signal will be treated as noise and will be attenuated. This results in compression and an artefact called 'glugging'.

  ***LEVEL may be adjusted as follows:***

  - Press F2 to select the LEVEL control. A box will appear around the numerical display to indicate that the control is selected.

  - Rotate the □-dial clockwise and/or anti-clockwise to alter LEVEL in steps of 0.01.

  - Rotating the □-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which LEVEL changes.

**ATTEN:**

ATTEN sets a maximum limit on the amount of noise that the DH-2 will remove at any time at any given frequency. It is quantified in dBs.

The side-effects introduced by inappropriate use of LEVEL are exaggerated when the DH-2 is asked to remove too much noise. Correct use of ATTEN will ensure that none of the side-effects caused by over- or under- processing are heard in the output signal. In cases of difficulty, you can often trade off the side-effects against the amount of hiss reduction.

***ATTEN may be adjusted as follows:***

- Press F3 to select the ATTEN control. A box will appear around the numerical display to indicate that the control is selected.

- Rotate the □-dial clockwise and/or anti-clockwise to alter ATTEN in steps of 0.1.

- Rotating the □-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which ATTEN changes.

**BRT:**

Brightness controls how the DH-2 compromises between the various side-effects. Its theoretical optimum is 50. However, in practice, the optimum setting for Brightness depends upon the nature of the signal and the hiss contained within it. It may also be influenced by any compromises you wish to make in order to achieve certain results.

- Brightness = too low
  At lower brightness settings the DH-2 will favour consistent compression instead of more unmusical or artificial-sounding artefacts. Generally, a low brightness is appropriate if you want to remove a lot of noise.

- Brightness = too high
  At higher brightness settings the DH-2 will minimise the compression of any low-level signals, but becomes susceptible to twitters and noise pumping. A high brightness is suitable when you wish to remove a small amount of noise .

*Note: You will find that the threshold and the brightness interact. Higher brightnesses will require slightly higher thresholds than lower brightnesses.*

***BRT may be adjusted as follows:***

- Press F4 to select the BRT control. A box will appear around the numerical display to indicate that the control is selected.

- Rotate the □-dial clockwise and/or anti-clockwise to alter BRT in steps of 0.01.

- Rotating the □-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which BRT changes.

**Ganging:**

The DH-2 may be used to process stereo material. The left and right channels of such material can be entirely independent and exhibit quite different noise characteristics. The “Ganging” control allows you to select which channel(s) are affected when you adjust LEVEL, ATTEN, and BRT.

The Ganging control has three modes:

Ganged: Adjusting the controls affects the left and right channels identically unless such adjustment would move a channel beyond the limits of the scale. In this mode, the numeric readouts beneath the control bars displays the average value of the left and right channels’ values.

Left: Only the left channel is affected, and the numeric readouts beneath the control bars displays the left channel’s values.

Right: Only the right channel is affected, and the numeric readouts beneath the control bars displays the right channel's values.

Press F1 to toggle between modes.

**Clear:**

The settings of LEVEL, ATTEN, BRT, and Ganging Controls may be returned to their default values simply by pressing CLEAR.

No other DH-2 controls or options are affected by this operation.

## PAGE 2: INPUT/OUTPUT CONTROL PAGE (I/O CONTROL)

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the I/O CONTROL PAGE appears.

This page allows you to determine the input used, the sampling frequency of the Analogue to Digital Converters, the digital input error detection level, and the digital output format.

(Remember that all outputs are permanently active, and that they do not require selecting, but that the same digital data is supplied to both AES/EBU and SPDIF outputs. The data format will therefore only be appropriate for one digital output at any given time.)

There are three options in the I/O Control Page:

### A. DIGITAL OUTPUT:

This option defaults to AES/EBU. To toggle between the two output modes, AES/EBU and SPDIF, press the Function Key marked 'A' on the LCD screen.

- AES/EBU FORMAT:

  When AES/EBU is selected, both the phono and XLR connectors will carry AES/EBU specification audio data. You should patch the output from the XLR connectors to your recording device.

  The DH-2 features 24-bit input and output resolution when AES/EBU is selected.

- SPDIF FORMAT:

  When SPDIF is selected, both the phono and XLR connectors will carry SPDIF specification audio data. You should patch the output from the phono connectors to your recording device.

  TPDF dithering will be applied to the digital data at the 16-bit level and is always applied at the analogue output.

## B. INPUT SOURCE:

There are three input sources: AES/EBU, SPDIF and ANALOGUE.

To toggle between the input sources press the Function Key marked 'B' on the LCD screen. The Status LEDs will indicate the inputs selected and the sample rate received (digital) or selected for conversion (analogue).

- SAMPLE RATE OF INCOMING DIGITAL SIGNAL:

  When the DH-2 is switched to receive digital audio data, the 'DIGITAL' LED will be lit, and the front panel LEDs will indicate the sample rate of the digital signal presented to the inputs:

| | | |
|---|---|---|
| neither 44.1 nor 48 kHz LED lit | = | 32 kHz signal presented to inputs |
| 44.1 kHz LED lit | = | 44.1 kHz signal presented to inputs |
| 48 kHz LED lit | = | 48 kHz signal presented to inputs |

- CLOCK DETECTION:

  If the DH-2 fails to detect a digital signal within the following limits, the 44.1kHz and 48kHz LEDs will flash continually. This will be irrespective of any other system settings.

| | | | |
|---|---|---|---|
| Acceptable ranges: | 44.1kHz | ± | 4% |
| | 48kHz | ± | 4% |
| | 32kHz | ± | 4% |

- SAMPLE RATE OF A TO D CONVERTERS

  When the DH-2 is switched to receive analogue audio data, the 'DIGITAL' LED will not be lit, and the front panel LEDs will indicate the sample rate of the analogue-to-digital converters.

  The ADCs in the DH-2 do not offer a 32kHz option unless synchronised to an external 32kHz source.

## C. A TO D FREQUENCY (INPUT SOURCE = ANALOGUE)

The ADC frequency may be selected by two, fundamentally different, methods. The first is to select one of the internal clock frequencies available, the second is to control the sample rate by using an external clock.

- INTERNAL CLOCK FREQUENCIES

  To toggle between the DH-2’s internal 44.1kHz and 48kHz sampling frequencies (and between AES Sync and SPDIF Sync - see below) press the Function Key marked 'C' on the LCD screen. The change in frequency will be shown on-screen and also by the Status LEDs.

  *Note: The sampling frequency reverts to 44.1kHz on Coldstart.*

- EXTERNAL SYNCHRONISATION

  The DH-2 clock may be synchronised to either the AES/EBU input or the SPDIF input. Connecting a valid digital input to either of these and selecting AES Sync or SPDIF Sync (as appropriate) will lock the DH-2 to the external clock.

  If the external clock falls within the acceptable ranges of each of the standard sample rates (44.1kHz, 48kHz, 32kHz) the clock frequency will be shown on the LEDs. If the external clock lies outside these ranges the DH-2 will still function, and good audio will be produced at the analogue output. Whether the digital output will be usable will then be determined by the flexibility of other devices in the digital audio chain.

  To toggle between AES Sync and SPDIF Sync (and also between the internal 44.1kHz and 48kHz sampling frequencies) press the Function Key marked 'C' on the LCD screen.

  *Note: If external synchronisation is requested, but no valid signal is detected at the appropriate digital input, the DIGITAL LED will flash to indicate the error.*

## D. RECEIVER ERROR LEVEL (INPUT SOURCE = AES/EBU or SPDIF)

The DH-2 features sophisticated software which detects and analyses both fatal and non-fatal errors in the incoming digital audio data.

You may select one of four error levels which will cause the front panel 'DIGITAL' LED to flash if the incoming data contains an error equal to or worse than the selected level.

The error levels are:

- **1 - Lock**

  This is the 'weakest' detector and will only cause the LED to flash when the DH-2 believes that there is no usable signal being presented to the selected digital input.

- **2 - Code**

  If there is an incoming signal yet the LED flashes on error level 2, the DH-2 is indicating that the signal contains coding violations. In some cases you may obtain usable audio. However, this warning may be caused by non-AES/EBU or non-SPDIF data being presented. In these cases any audio produced will almost certainly be unusable.

- **3 - Trans**

  This indicates that the incoming digital audio data is of poor quality (i.e very noisy or jittery) and that undetectable data errors are likely. These errors will not be corrected by any standard AES/EBU or SPDIF device and may lead to audio degradations.

- **4 - Valid**

  This is the most stringent test of the incoming data, and will cause the LED to flash if the DH-2 determines that any of the data contained in the signal is not valid. This is often non-fatal (i.e. you will hear perfectly good audio) but it indicates that some device or anomaly in your audio chain is generating digital audio data outside of the AES/EBU or SPDIF specifications published by their respective bodies. Please note however that, if the digital LED does not flash, this can not be taken as an absolute statement that the signal conforms to specification.

*Note: If the error level selected detects an error, the digital audio signal will be coded as INVALID by the DH-2. Many manufacturers' devices do not recognise or act upon this code, but those that do may refuse to accept or record the audio.*

# PAGE 3: REMOTE CONTROL

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the REMOTE CONTROL PAGE appears.

The DH-2 features intelligent 'auto-detection' software which monitors the RS232, MIDI, and SMPTE/EBU (if fitted) inputs and responds to data received on each and any of them. This eliminates the need for a control to select the remote control to be used.

It is only necessary, therefore, to select the Channel on which the DH-2 receives commands over MIDI.

## MIDI

CEDAR Audio Ltd does not produce software for remote devices to control the DH-2 over MIDI.

- MIDI CHANNEL

  Ensure that button A is highlighted by a box. It is then possible to change the MIDI Channel turn the □-dial clockwise (to increase) or anti-clockwise (to decrease) the MIDI Channel.

  To toggle this function on/off press the Function Key marked 'A'.
  On Coldstart the MIDI Channel defaults to 1.

## RS232

CEDAR Audio Ltd does not produce software for remote devices to control the DH-2 over RS232. However, for users wishing to implement their own control software, the RS232 Protocol is outlined in the chapter 'RS232 Protocol'.

## SMPTE/EBU Timecode

A separate SMPTE/EBU reader/generator board may be purchased and fitted inside your DH-2. Please contact your dealer for details of this.

## PAGE 4: STATUS PAGE

Access the STATUS PAGE by holding down Function Key F5 and then pressing the Dedicated Function Key PAGE.

Should the DH-2 fail to function, or appear to function incorrectly, there may be an error contained within the digital audio data received at the System's inputs. The Receiver Error Level (see above) will notify you when an error has occurred, but it will not tell you what it is. For many users, this information will be adequate, but the DH-2 is capable of reporting errors and other status information in more detail.

The STATUS PAGE will give you information regarding the current status of the DH-2, and will give you details regarding any errors which have occurred since the unit was switched on.

Three items of information will always be reported by the DH-2. These are:

| | | | |
|---|---|---|---|
| • | DSP1: | Status | Crashed / Timed Out / Running |
| • | DSP2: | Status | Crashed / Timed Out / Running |
| • | I/O: | Condition | Error / Emphasis, Sample Rate |

If a remote control error is detected, a fourth field will appear:

| | | | |
|---|---|---|---|
| • | Comms: | Error | Illegal Checkbyte / Illegal Command Size |

### STATUS INDICATORS

The front panel LEDs will help to identify the possible cause if the unit fails to function. The following table lists all possible combinations of LED error indications:

| **LED flashing:** | **Condition:** |
|---|---|
| Digital | The digital input violates the Receiver Error Level<br>*or* no digital sync is present (if requested in I/O page) |
| 44.1 *and* 48kHz | Unknown sample rate received at inputs |
| Bypass/Pre/Post | One or both of the DSPs have crashed. |

## STATUS PAGE DEFINITIONS:

| | |
|---|---|
| Crashed | The DH-2 DSPs are failing to function. The only recourse is to switch the unit off, wait for a few seconds, and then switch on again. If this error re-occurs please refer your DH-2 to an authorised service centre. |
| Timed Out | If, for any reason, the DH-2 drops out of real-time (fails to pass audio to the output) this error will be reported. This should only occur if a sample rate of greater than 50kHz is presented to one of the digital inputs. This error is non-fatal, and the DH-2 should continue to function normally after it has occurred. |
| Running | The DH-2 DSPs are functioning correctly and, moreover, have been doing so since the unit was switched on. |
| Error | If the DIGITAL LED is flashing the most serious error will be detailed at this point. Errors are fully detailed in the DH-2 Service Manual. |
| Emphasis | If no error is detected, the I/O status will display the Emphasis condition:<br><br>• OFF<br><br>The Emphasis bit is not set.<br>The DAC de-emphasis will not be engaged.<br><br>• 50/15<br><br>The Emphasis bit is set to 50/15 µS.<br>The DAC de-emphasis will be engaged.<br><br>• J17 (AES/EBU only)<br><br>The Emphasis bit is set to CCITT J17.<br>The DAC de-emphasis will not be engaged.<br><br>• Unknown (AES/EBU only)<br><br>The Emphasis status is not indicated.<br>The DAC emphasis status will not be altered. |
| Sample Rate | If no digital data error is detected, the measured sample rate presented to the digital inputs will be displayed to the nearest 100Hz. |
| Illegal Checkbyte | The RS232 or MIDI has received a command packet containing an illegal checkbyte (byte2). |

| | |
|---|---|
| Illegal Command Type | The RS232 or MIDI has received a command packet containing an illegal command type (byte4). |

# TUTORIAL

One method for determining the correct values of the DH-2 noise removal controls is outlined below.

It is important that the audio presented to the DH-2 is free of clicks and crackle. This is because these degradations will interfere with the de-hissing process and prevent you from reaching an optimal result.

Please note that the tutorial assumes that the material is stereo and exhibits virtually identical noise characteristics in each channel.

1. Ensure that the DH-2 is in POST and that BYPASS is OFF.

2. Select the Control Page and press CLEAR to reset the values of the DH-2's process controls to their defaults.

3. Your first task will be to find the most appropriate setting for the LEVEL control. This will be the single biggest influence on the quality of the processed signal.

   Starting with LEVEL at 0.00, use the □-dial to increase the value. First you will notice that very little happens. At these levels you will be in a region in which instruments sound generally uncompressed, although there may be noise artefacts present, and the high frequency content may occasionally sound gated. You might also notice the side-effect known as 'twittering'.

   At some point as you continue to increase LEVEL, the amount of hiss will begin to decrease rapidly. The point at which this occurs is defined by the nature of the noise contained within the signal. With further increases, the 'twittering' will disappear, but you will then cause the on-set of high frequency compression, gating, and the side-effect sometimes called 'glugging'.

   The optimum value of value of LEVEL is around the crossover point between the twitters and the glugging. Unfortunately, if the hiss is highly inconsistent, there may be an overlap between the 'twitter' and the 'glugging' regions, and no setting that can be considered optimum.

   *Note: If you are using the analogue inputs you must not adjust the front panel INPUT LEVEL CONTROL after you have found a suitable LEVEL. Such adjustment will, of course, alter the amount of noise being presented to the DH-2's processors, and make the initial LEVEL inaccurate.*

4. Some users prefer to adjust BRT next, whilst others adjust ATTEN. Ultimately, you will discover that the three main controls are closely related, and that, for best results, you will need to adjust each of them a number of times. However, in this tutorial we will adjust BRT next, followed by ATTEN.

5. If you want a particularly clean signal, or are processing material with 'grungy' hiss, you will need to set the Brightness to a low value. Similarly, if you want a to achieve an acceptable result quickly, use a low Brightness.

   If your defining criterion is signal quality, a middle Brightness (together with a moderate attenuation) will usually offer best results.

   If you want a very clean signal and are prepared to accept the risk of low-level artefacts, you will obtain the best results using a high value for Brightness.

*Note: You may wish to adjust the LEVEL while you are determining the most suitable setting for BRT. Where possible, listen to the fidelity of any background instruments, and bear in mind the general feeling of 'air' or 'presence' in the signal, as well as the quality of any residual noise. In general, the DH-2 is less sensitive to the Threshold setting when the value of Brightness is low than it is when the Brightness is high.*

6. You can now adjust the ATTEN control to determine the amount of noise removed.

   Increase ATTEN from -40.0 to 0.0, at which point you will hear that the processed signal is identical to the unprocessed. (This is because the ATTEN control is limiting the amount of noise removal to -0dB at every frequency - i.e. there is no effect.)

   You may now reduce ATTEN to a level defined by the material and your taste. However, you will notice that, if LEVEL is high, you can only reduce ATTEN by a few dBs before the on-set of side-effects such as loss of transients and loss of high frequencies. If LEVEL is low, ATTEN can be reduced further, but with reduced effect.

   As you reduce ATTEN you may notice one of two detrimental effects occurring:

   - If there were ‘twitters’ present after step (3), and if you reduce ATTEN beyond the optimal level for the specific material being processed, the twitters may be re-introduced as a form of high-frequency noise modulation.

   - If there was loss of high frequencies present after step (3), you will notice that this loss is gradually re-introduced as you decrease ATTEN.

7. It is unlikely that the values of the three controls are already optimised, so you should now return to step (3) and attempt to find a better value for LEVEL. Having done this you will, no doubt, wish to modify BRT and ATTEN further.

   Continued fine-tuning of these controls will lead to excellent noise removal with few or no side-effects. However, the DH-2 is not a magic wand, and it may not be possible to restore some (especially badly degraded) material beyond a certain point. Only experience will enable you to judge whether you have removed as much noise as possible without unacceptable consequences.

In general, if the value of the Attenuation is small (-6dB or less) the Threshold can be set lower and the Brightness can be set higher without introducing twitters or hiss modulation.

# THE TUTORIAL TAPE

We have supplied you with a DAT containing three samples of hissy music. These samples may prove useful in helping you to learn the features and capabilities of your DH-2.

### TRACK 1: SEGOVIA

Solo guitar is often difficult to restore, but this track is of generally good quality and does not have a particularly high level of hiss. It is, therefore, fairly simple to process and achieve excellent results. LEVEL does not need to be set very high (although high enough to avoid twittering) and this minimises the danger of unwanted side-effects. ATTEN should be set more carefully because an acoustic guitar offers a rich harmonic content that could be damaged by over-processing. You will find that BRT can be set close to its theoretical optimum.

| | | | | | | |
|---|---|---|---|---|---|---|
| Light processing: | LEVEL: | 29.4/19.8 | ATTEN: | -4/-3 | BRT: | 50.3/50.3 |
| Heavier processing: | LEVEL: | 29.4/19.8 | ATTEN: | -6/-5 | BRT: | 30.2/30.2 |

*Note: This is a stereo track, so values are shown as Left/Right pairs.*

### TRACK 2: LOUIS ARMSTRONG AND THE HOT FIVE

This track is taken from a 78rpm record and is therefore of lower quality, with limited frequency response and a higher level of hiss. It is, therefore, more difficult to process than track 1. Firstly, LEVEL must be set much higher than before otherwise the higher level of hiss will cause significant amounts of twittering. Similarly, ATTEN will require a greater value in order to reduce the hiss to an acceptable volume. In short, the track will require heavier processing because the noise is much more noticeable than on track 1.

The following settings are average values which will work reasonably well for the whole track:

| | | | | | | |
|---|---|---|---|---|---|---|
| Light processing: | LEVEL: | 40.9 | ATTEN: | -5 | BRT: | 50.8 |
| Heavier processing: | LEVEL: | 40.9 | ATTEN: | -7 | BRT: | 30.3 |

### TRACK 3: RICHARD GRESKO

This is a hissy tape recording from the 1970s. The quality is reasonably good but, like solo guitar, the solo piano can be very difficult to restore without unacceptable side-effects. However, and unlike track 1, the hiss is mostly constant, so a lower LEVEL may be used without introducing twittering. With mild ATTEN and typical values for BRT, a very satisfactory result may be obtained.

| | | | | | | |
|---|---|---|---|---|---|---|
| Light processing: | LEVEL: | 21.0 | ATTEN: | -3 | BRT: | 50.8 |
| Heavier processing: | LEVEL: | 21.0 | ATTEN: | -5 | BRT: | 30.8 |

# REMOTE CONTROL PROTOCOLS

## 1. RS232

RS232 is defined in the DH-2 as:

**9600 baud**
**8 bits data**
**1 stop bit**
**No parity**

A command packet contains 6 bytes. These are:

| | |
|---|---|
| byte 1: | channel number byte: must be 0xAF |
| byte 2: | Checkbyte. Fixed: must be 0x63 |
| byte 3: | command number (see below) |
| byte 4: | Command type. Fixed: 0x07 |
| byte 5: | command value HIGH byte |
| byte 6: | command value LOW byte |

The HIGH and LOW bytes together form a signed integer.

| **Command Numbers:** | | **Command Values:** | | |
|---|---|---|---|---|
| 0xF7 | Clear Errors command | Any value | = | Clear all error messages |
| 0xF8 | Select Page command | 1 | = | Control Page |
| | | 6 | = | I/O Control Page |
| | | 7 | = | Status Page |
| | | 15 | = | Remote Control Page |
| | | -1 | = | Toggle between Pages |
| | | Any other value | = | Refresh |
| 0xF9 | Pre/Post command | 0 | = | Pre |
| | | 1 | = | Post |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |
| 0xFA | Bypass command | 0 | = | Bypass OFF |
| | | 1 | = | Bypass ON |
| | | 2 | = | RESERVED VALUE |
| | | 3 | = | RESERVED VALUE |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |
| 0xC0 | Digital Output Format | 0x80 | = | SPDIF |
| | | 0x00 | = | AES/EBU |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |

| | | | | |
|---|---|---|---|---|
| 0xC1 | Input Source | 0 | = | Analogue |
| | | 1 | = | SPDIF |
| | | 2 | = | AES/EBU |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |
| 0xC2 | A to D Frequency | 0 | = | 44.1kHz |
| | | 1 | = | 48kHz |
| | | 2 | = | SPDIF Sync |
| | | 3 | = | AES/EBU Sync |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |
| 0xC3 | Receiver Error Level | 0 | = | 1 - Lock |
| | | 1 | = | 2 - Code |
| | | 2 | = | 3 - Trans |
| | | 3 | = | 4 - Valid |
| | | -1 | = | Toggle |
| | | Any other value | = | Refresh |
| 0x20 | Set Left LEVEL | Any value | = | (Left LEVEL) x 100 |
| 0x30 | Alter Left LEVEL | Any value | = | ∆ (Left LEVEL) x 100 |
| 0x21 | Set Right LEVEL | Any value | = | (Right LEVEL) x 100 |
| 0x31 | Alter Right LEVEL | Any value | = | ∆ (Right LEVEL) x 100 |
| 0x22 | Set Left ATTEN | Any value | = | (Left ATTEN) x 100 |
| 0x32 | Alter Left ATTEN | Any value | = | ∆ (Left ATTEN) x 100 |
| 0x23 | Set Right ATTEN | Any value | = | (Right ATTEN) x 100 |
| 0x33 | Alter Right ATTEN | Any value | = | ∆ (Right ATTEN) x 100 |
| 0x24 | Set Left BRT | Any value | = | (Left BRT) x 100 |
| 0x34 | Alter Left BRT | Any value | = | ∆ (Left BRT) x 100 |
| 0x25 | Set Right BRT | Any value | = | (Right BRT) x 100 |
| 0x35 | Alter Right BRT | Any value | = | ∆ (Right BRT) x 100 |

## 2. MIDI

The DH-2 is permanently set to transmit any change of control page parameters or Pre/Post state via MIDI except when such a change is initiated by an RS232 or MIDI command. Therefore, if a MIDI sequencer such as Cubase™, Notator™, or EditTrack™ is connected to the DH-2, it will receive a running history of the unit's operation.

If your sequencer and audio sources are able to send and receive timecode, then the DH-2's MIDI capability may be used as the basis for an automation system.

*Note: The absolute parameter values are not transmitted or received, so the user must ensure that any changes are relative to a desired starting value which can be set using MIDI DUMP.*

*If a MIDI DUMP of all control page parameters and the Pre/Post state is required, pressing ENTER at any time will initiate the DUMP.*

### Additional MIDI Command

The DH-2 will receive LOCAL ON and LOCAL OFF commands.
The Status Page will notify you of the current state.
Both WARMSTART and COLDSTART always set LOCAL ON.

This command cannot be initiated from the front panel of the DH-2.

# SELF TEST MODE

The DH-2 features a powerful self-test mode which enables the System to check the operation of each of its major sub-systems, plus all of the user controls.

**To enter the self-test mode:**
Switch on the DH-2 while holding down the ENTER key. The DH-2 will perform each test in turn, and you may move to the next test by pressing the ENTER key. Consequently, any test may be skipped by pressing the ENTER key.

*Note: Whilst the SELF-TEST is in progress, the ENTER key will not initiate a MIDI DUMP.*

## ROUTINE 1: BUTTON TESTING ROUTINE

The DH-2 will invite you to press each of the Function Keys (except ENTER) and each of the Dedicated Function Keys. Pressing a key will cause the display to change from OFF to ON.

## ROUTINE 2: ATTENUATION KNOB TEST

The DH-2 will invite you to turn the Attenuation knob to check that the value displayed on screen matches the position of the knob..

## ROUTINE 3: □-dial (SPIN WHEEL) TEST

Rotate the □-dial to check that values change smoothly in both positive (clockwise) and negative (anti-clockwise) directions.

## ROUTINE 4: LED TEST

The DH-2 will flash all six green LEDs.

## ROUTINE 5: METER TEST

The DH-2 will invite you to turn the □-dial to vary the levels displayed by each of the four input and output meters in turn. Press ENTER to step to the next meter.

## ROUTINE 6: DSP1 TEST

The DH-2 will test its DSPs and internal memory. Please wait for this test to complete.

- If the System is fully functional the screen will display the message: **"Memory passed".**
- If a memory error is detected the screen will display the message: **"Memory error at: .....".**
- If a DSP failure is detected the screen will display the message: **"DSP1 is not responding".**

If you observe this message please repeat the self-test. If the message recurs please contact your dealer for assistance.

*WARNING: The DH-2 contains no user-serviceable parts. DO NOT UNDER ANY CIRCUMSTANCES attempt to service your unit.*

## ROUTINE 7: DSP2 TEST

As above.

## TEST COMPETED

Your DH-2 will now prompt you to press ENTER one more time to return you to operating mode (whether all tests have been passed or not).

Some failures will not stop you from using the DH-2 successfully. However, consistent failures should be notified to your dealer or directly to CEDAR Audio Ltd.

## A FINAL NOTE…

Some users use commercial test CDs or signal generators to test the operation of components within their audio systems. The DH-2 will generate distortion if a digital FSD (full-scale deflection) sine wave is applied to either the AES/EBU or SPDIF inputs. This does not imply that your DH-2 is faulty, and the effect should be ignored.

# CEDAR DH-2

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